1   <u>TITLE OF THE INVENTION</u>

2   Acoustic Echo Suppressor for Hands-free Speech Communication

3   <u>BACKGROUND OF THE INVENTION</u>

4   <u>Field of the Invention</u>

5   The present invention relates to an echo suppressor for application to

6   full-duplex speech communication such as hands-free telephony and voice

7   recognition in a noisy environment. The present invention is particularly

8   useful for applications where the acoustic echo path of a full-duplex speech

9   communication system is severely affected by nonlinear distortion.

10   <u>Description of the Related Art</u>

11   In a full-duplex speech communication system such as a telephone or a

12   notebook computer operating in a hands-free mode, distant signal from a far-

13   end talker is transmitted from the loudspeaker 2 and some of the acoustic

14   energy is sensed by the microphone 1 (Fig. 1). Acoustic echoes occur as a

15   result of the distant signal from the loudspeaker 2 being coupled through a

16   channel known as acoustic echo path to the microphone. The acoustically

17   coupled distant signal is then coupled into the return path and propagates

18   through the network to the far-end talker, giving an impression of an echo of

19   the talker's voice. In order to cancel the echo, a linear echo canceller 3 is

20   provided. As described in a technical paper "The hands-free telephone

21   problem: an annotated bibliography updated", Eberhard Hansler, Annals of

22   Telecommunications, 1994, pages 360-367, the linear echo canceller 3 has a

23   replica of the transfer function of the acoustic echo path to produce an echo

24   replica of distant signal. The echo replica is used in a subtractor 4, or residual

25   echo detector to cancel the echo contained in the output of microphone 1,

26   producing an echo-free local signal. A speech detector 5 is provided to

27   monitor the outputs of echo canceller 3 and subtractor 4 as well as the local

28   and distant signals for detecting speech activity of the near-end talker.

29   Speech detector 5 produces a zero or a near-zero output when the near-end

30   speech activity is high and a high-level output when it is low or zero.

1   The linear echo canceller 3 includes a linear adaptive filter 7. This filter

2   performs a linear filtering on the distant signal and supplies its output to the

3   subtractor 4, while its filter coefficients are constantly updated through a

4   feedback loop according to the output of subtractor 4. The updating

5   algorithm of linear adaptive filter 7 is a process of correlation calculation such

6   that the residual echo at the output of subtractor 4 is reduced to a minimum.

7   As a result, those components of the microphone signal which are correlated

8   with the distant signal are minimized. A multiplier 8 is provided in the

9   feedback loop to prevent near-end speech activity from disturbing the filter

10   coefficients. When the near-end speech activity is high, the output of speech

11   detector 5 is zero or near-zero, which nullifies the multiplier 8 so that the

12   filter coefficients are frozen.

13   Nonlinearity is of another concern to the design of the echo canceller.

14   The prior art echo cancellation may be satisfactory in so far as the

15   nonlinearity of the acoustic echo path is of small magnitude and the linear

16   echo canceller is able to replicate it. However, in practical systems the

17   operating characteristics of transducer elements of the loudspeaker are far

18   from ideal. Their nonlinear characteristics are of such a magnitude that the

19   linear echo canceller cannot completely replicate the transfer function of

20   nonlinear acoustic echo path. This is particularly true to cellular phones or

21   notebook computers where their small-sized loudspeakers are operated in a

22   high-powered hands-free mode. Due to their severe nonlinear characteristics,

23   acoustic echo cannot be suppressed by more than 20 dB. The remaining echo

24   component would propagate through the network and seriously impede the

25   distant talker.

26   SUMMARY OF THE INVENTION

27   It is therefore an object of the present invention to provide echo

28   suppression when the acoustic echo path of a full-duplex speech

29   communication system suffers severe nonlinear distortion resulting from

30   nonlinear operating characteristics of a loudspeaker.

1   According to a first aspect of the present invention, there is provided a

2   speech communication apparatus comprising a signal output transducer for

3   receiving a distant signal from a far-end talker and producing acoustic energy

4   of the distant signal, a signal input transducer for producing a near-end

5   signal which may contain a component representing a speech activity of a

6   near-end talker or an acoustic echo component, or both, wherein the acoustic

7   echo component occurs as a result of the distant signal being transmitted

8   through an acoustic echo path from the signal output transducer to the signal

9   input transducer, an echo canceller for producing an echo replica from the

10   distant signal and a residual echo, and a residual echo detector for detecting a

11   difference between the near-end signal and the echo replica and supplying

12   the difference as the residual echo to the echo canceller. A spectral shaper is

13   provided for receiving one of the near-end signal and the residual echo as a

14   first input signal, receiving the echo replica as a second input signal,

15   estimating from the first and second input signals the acoustic echo

16   component when the speech activity is low or zero, and shaping the spectrum

17   of the first input signal with the estimated acoustic echo component.

18   According to a second aspect of the present invention, there is

19   provided a speech communication apparatus comprising a signal output

20   transducer for receiving a distant signal from a far-end talker and producing

21   acoustic energy of the distant signal, means for dividing the distant signal

22   into a first set of subband frequency component signals, a signal input

23   transducer for producing a near-end signal which may contain a component

24   representing a speech activity of a near-end talker or an acoustic echo

25   component, or both, wherein the acoustic echo component occurs as a result

26   of the distant signal being transmitted through an acoustic echo path from the

27   signal output transducer to the signal input transducer, means for dividing

28   the near-end signal into a second set of subband frequency component

29   signals, and a plurality of subband echo suppressors. Each subband echo

30   suppressor comprises an echo canceller for producing an echo replica from a

1   corresponding one of the first set of subband frequency component signals

2   and a subband residual echo, a residual echo detector for detecting a

3   difference between a corresponding one of the second set of subband

4   frequency component signals and the echo replica and supplying the

5   difference as the residual echo to the echo canceller, and subband spectral

6   shaping means for receiving the residual echo as a first subband input signal,

7   receiving the echo replica as a second subband input signal, estimating from

8   the first and second input signals the acoustic echo component when the

9   speech activity is low or zero, and shaping the first subband input signal with

10 the estimated acoustic echo component to produce an output signal of the

11 subband echo suppressor. The output signals of the plurality of subband

12 echo suppressors are combined together to produce a local signal for

13 transmission to the far-end talker.

14   BRIEF DESCRIPTION OF THE DRAWINGS

15   The present invention will be described in detail further with reference

16 to the following drawings, in which:

17   Fig. 1 is a block diagram of a prior art echo suppressor;

18   Fig. 2 is a block diagram of an echo suppressor according to a first

19 embodiment of the present invention;

20   Fig. 3 is a block diagram of the spectral shaper of Fig. 2 implemented

21 in a spectral subtractor according to a first embodiment of the present

22 invention;

23   Fig. 4 is a block diagram of one form of the Fourier coefficient

24 subtractors of Fig. 3;

25   Figs. 5A and 5B are block diagrams of modified forms of the Fourier

26 coefficient subtractors;

27   Fig. 6 is a block diagram of a second embodiment of the spectral

28 shaper of Fig. 2 implemented in a spectral suppressor;

29   Fig. 7 is a block diagram of one form of the Fourier coefficient

30 multipliers of Fig. 6;

1   Fig. 8 is a block diagram of a modified form of the Fourier coefficient

2 multipliers;

3   Fig. 9 is a block diagram of an echo suppressor according to a modified

4 embodiment of the present invention;

5   Fig. 10 is a block diagram of an echo suppressor according to a further

6 modification of the present invention;

7   Fig. 11 is a block diagram of the echo canceller of Fig. 10;

8   Fig. 12 is a block diagram of the spectral subtractor of Fig. 10; and

9   Fig. 13 is a block diagram of an echo suppressor according to a third

10 embodiment of the present invention.

11   DETAILED DESCRIPTION

12   Referring now to Fig. 2, there is shown a hands-free speech

13 communication system, or echo suppressor according to the present

14 invention in which parts corresponding in significance to those in Fig. 1 are

15 marked with the same numerals and the description thereof is omitted for

16 simplicity. The echo suppressor may be coupled to a two-wire subscriber line

17 using a well-known technique such as time compression multiplexing (TCM)

18 or hybrid coupling (two-wire four-wire conversion). The echo suppressor

19 includes a spectral shaper 10, which is configured to receive a first input

20 signal either from subtractor 4 or microphone 1 and a second input signal

21 from linear echo canceller 3. Spectral shaper 10 further receives the output of

22 speech detector 5 as a disabling signal to nullify its operation when near-end

23 speech activity is high.

24   If the output of subtractor 4 is used as an input to the spectral shaper,

25 the echo cancellation is performed primarily in the subtractor 4. If the output

26 of microphone 1 is used instead, the spectral shaper 10 takes the

27 responsibility for the cancellation of acoustic echoes. For the purpose of

28 disclosure, the spectral shaper uses the output of subtractor 4 as its first input

29 signal.

30   According to a first embodiment of the present invention, the spectral

1   shaper 10 is implemented in a spectral subtractor as shown in Fig. 3. In the

2   spectral subtractor 10, the first input signal is divided into a first set of

3   subband frequency components and the second input signal is likewise

4   divided into a second set of subband frequency components. From both sets

5   of subband frequency components, nonlinear subband echo components are

6   estimated to produce a set of echo cancellation signals respectively

7   corresponding to the subband frequencies. Nonlinear echo components

8   respectively contained in the first set of subband frequency components are

9   cancelled by the corresponding subband echo cancellation signals and then

10 combined together into a local signal for transmission.

11   In Fig. 3, the spectral shaper is configured as a spectral subtractor

12 which includes Fourier transform converters (or spectral splitters) 11 and 12.

13 Fourier transform converter 11 performs M-point Fourier transform

14 calculations on its input signal from the subtractor 4 on a sample-by-sample

15 basis to produce a first set of Fourier coefficients as the results of the

16 calculations. As a result, the spectrum of the output signal of subtractor 4 is

17 split into transform-domain subband frequency components S1, S2, ..., Sm

18 corresponding to the Fourier coefficients of the first set. In like manner,

19 Fourier transform converter 12 performs M-point Fourier transform

20 calculations on its input signal from the echo canceller 3. The results of the

21 calculations are produced as a second set of Fourier coefficients

22 corresponding to subband frequency components R1, R2, ..., Rm.

23   A plurality of Fourier coefficient subtractors 13-1 through 13-m are

24 provided. Each Fourier coefficient subtractor 13-i has a pair of input

25 terminals for respectively receiving a subband component Si from the Fourier

26 transform converter 11 and a subband component Ri from the Fourier

27 transform converter 12.

28   The output of speech detector 5 is supplied to all Fourier coefficient

29 subtractors 13 to control their time constant values (smoothing coefficients).

30 As described in detail later, the estimate of each subband echo component is

1 represented by an average value of the ratio of power or amplitude of its first

2 input signal of each Fourier coefficient subtractor 13 to power or amplitude of

3 its second input signal. Preferably, the time constant used to average these

4 input signals is controlled such that it is smaller when the input signals are

5 increasing than it is when they are decreasing. Alternatively, the averaging

6 time constant value is long or infinite when speech activity is high and is

7 short when speech activity is low. Additionally, when speech activity is low

8 the averaging time constant value is smaller when the ratio is increasing than

9 it is when the ratio is decreasing.

10   The output signals of all Fourier coefficient subtractors 13 are

11 combined together in an inverse Fourier transform converter (spectral

12 combiner) 14. Converter 14 performs an inverse Fourier transform

13 calculation on each of its input signals and the real parts of the results of

14 calculation are combined together to be delivered as a local signal.

15   Details of each Fourier coefficient subtractor 13i are shown in Fig. 4.

16 Fourier coefficient subtractor 13i comprises absolute value circuits 15A and

17 15B. Absolute value circuits 15A and 15B receive the subband components Si

18 and Ri from the Fourier transform converters 11 and 12 and produce absolute

19 values of the Fourier coefficients Si and Ri, respectively. A ratio of the

20 absolute value Si to the absolute value Ri is obtained by a divider 16. The

21 output of divider 16 is coupled to a smoother 17. The smoother 17 produces

22 an average value of the Si/Ri ratio with a varying time constant depending

23 on the level of speech activity detected by the speech detector 5. When near-

24 end speech activity is high, the time constant is set to a large or infinite value.

25 When the speech activity is low or zero, the smoother 17 operates with a

26 small time constant value. Smoother 17 has the effect of stabilizing the

27 Fourier coefficient subtractor 13 when the ratio varies violently in response to

28 a high near-end speech signal so that a reliable output signal is obtained.

29   More specifically, the smoother 17 may be implemented in a leaky

30 integrator or a first-order IIR (infinite impulse response) lowpass filter. If the

1 smoother is implemented in a leaky integrator, it is comprised of a subtractor

2 21, a multiplier 22, an adder 23, a limiter 24, and a one-sample delay element

3 25, all of which are connected in a closed loop by coupling the output of the

4 delay 25 to the subtractor 21 and adder 23. The ratio output of divider 16 is

5 supplied to the subtractor 21 where the output of delay 25 is subtracted from

6 the Si/Ri ratio. The output of subtractor 21 is supplied to the multiplier 22,

7 where the ratio is scaled in accordance with a scale factor supplied from a

8 smoothing coefficient (time constant) selector 26. Selector 26 responds to the

9 output of speech detector 5 by setting the scale factor to a very small non-

10 negative coefficient "0.0", for example, when speech activity is low or zero so

11 that the smoothing time constant remains unchanged. When speech activity

12 is high, the selector 26 sets the scale factor to a relatively high positive

13 coefficient "0.005". The output of multiplier 22 is summed with the output of

14 delay 25 to produce a sum signal. After limiting its amplitude to within

15 upper and lower bounds by the limiter 24, the sum signal is fed to the delay

16 25, so that the scaled ratio is integrated over time and averaged out when the

17 near-end talker is in a talking mode.

18   Preferably, the smoothing coefficient selector 26 is modified so that it

19 responds to the output of subtractor 21, as indicated by a broken line 27, in

20 addition to the output of speech detector 5. When the near-end speech

21 activity is high, the selector 26 supplies a non-negative coefficient of 0.0, for

22 example. When the near-end speech activity is low or zero, the selector 26

23 supplies a relatively large smoothing coefficient of 0.01, for example, if the

24 output of subtractor 21 is positive (indicating that it is producing an

25 increasing output) and a relatively small coefficient of 0.001 if the subtractor

26 21 is producing a negative output (indicating that the output is decreasing).

27 With these time-varying smoothing scale factors, the output of the smoother

28 17 varies sharply at the rising edge of a transition and varies slowly at the

29 falling edge of the transition. With this arrangement, the output of the

30 smoother 17 is made to follow the sharply rising and gradually falling edges

1 of natural sounds. Acoustic echo can be estimated with a higher degree of

2 accuracy.

3   The ratio averaged by smoother 17 is withdrawn from the output of

4 limiter 24 and multiplied in a multiplier 18 by the signal Ri.

5   Since the Si/Ri ratio obtained by the divider 16 can be considered as a

6 quotient of the Fourier coefficient of a near-end subband component divided

7 by the Fourier coefficient of a far-end subband component, multiplying the

8 ratio by the Fourier coefficient of subband component Ri of the echo replica

9 in the multiplier 18 results in a value which is equal to the Fourier coefficient

10 of the subband component "i" of the near-end signal and represents an

11 estimate of the Fourier coefficient of the residual subband component in the

12 echo replica. The estimated residual subband component of the echo replica

13 obtained by the multiplier 18 is supplied to a subtractor 19 to cancel the

14 acoustic subband echo component contained in the near-end speech signal Si.

15   It is seen that the spectral subtractor 10 performs nonlinear

16 calculations in the frequency domain. In this respect, the timing variations of

17 subband components are of important consideration. Nonlinear distortion of

18 the echo channel in the acoustic path is effectively compensated for by

19 adaptively adjusting the timings of the subband signals. In the time domain,

20 the linear echo canceller 3 performs this operation in a manner

21 complementarily to the operation of spectral subtractor 10.

22   In quantitative terms, the operation of the spectral subtractor,

23 particularly, the Fourier coefficient subtractors 13 is analyzed as follows:

24   If the Fourier coefficient of the near-end signal is denoted as S, the

25 following relation holds:

26      $S = A + E + N$ (1)

27 where, A is the near-end talker's speech component, E is an echo component

28 and N is a noise component. The Fourier coefficient of the far-end signal (R)

29 is in phase with the Fourier coefficient S of the near-end signal. If A is not

30 present, i.e., there is no near-end speech activity, the near-end signal S is E +

1   N, which can be completely discarded. Under such conditions, the following

2   relation holds:

3   $\qquad P1 = Av\,[S/R] = Av\,[(E + N)/R]$ (2)

4   where, P1 is the output of the smoother 17 and Av [·] represents a smoothing

5   operator. P1 approximates the proportion of the distant signal R that

6   contributes to the echo and can be treated as the "echo gain" of an acoustic

7   echo path.

8   $\qquad$ If the output of multiplier 18 is denoted as P2, the following relation is

9   established:

10 $\qquad P2 = P1 \times R$

11 $\qquad\qquad = R \times Av\,[(E + N)/R]$

12 $\qquad\qquad = Ex\,[E + N]$ (3)

13 where, Ex [·] represents an estimate.

14 $\qquad$ The output signal P3 of subtractor 19 is given as follows:

15 $\qquad P3 = S - P2$

16 $\qquad\qquad = S - (R \times Av\,[(E + N)/R])$ (4a)

17 $\qquad\qquad = (A + E + N) - Ex\,[E + N])$ (4b)

18 $\qquad\qquad = Ex\,[A]$ (4c)

19 Equation (4c) implies that an estimate of a noiseless, echo-free speech

20 component A can be obtained for the near-end Fourier coefficient S.

21 $\qquad$ In addition, the nonlinear echo component of a distant signal

22 contained in the output signal of microphone 1 can be treated as harmonics of

23 the distant signal. Consider an echo component E by assuming that it

24 exclusively contains harmonics. Equations (2) and (3) show that, in principle,

25 the echo component E can be completely removed in so far as the distant

26 Fourier transform component R is non-zero. However, for cancelling the

27 echo component E, the accuracy of the echo gain P1 that can be attained is

28 also important. Since the amount of harmonics varies significantly from

29 instant to instant depending on the characteristics of the distant signal, such

30 as amplitude, it is desirable that the timing variation of the distant Fourier

1   transform component R be as synchronized as possible to the timing

2   variation of the echo component E, as indicated by the denominator and

3   nominator of Equation (2). High degree of accuracy can be obtained for the

4   echo gain P1 if the timing variations of these components occur in

5   synchronism to each other. It is an advantageous feature of the present

6   invention that, since the spectral subtractor derives the Fourier transform

7   component R from the output of linear echo canceller 3, the timing variations

8   of components E and R are substantially synchronized to each other.

9   Another feature of the present invention resides in the fact that even if

10   the linear echo canceller 3 makes an error in the echo path estimation,

11   resulting in a residual echo at the output of subtractor 4, the spectral

12   subtractor of this invention can remove such a residual echo. In this regard,

13   the above discussion also applies to the type of echo components E that

14   contain non-harmonics of the distant signal.

15   A further feature of the present invention is that the use of the spectral

16   subtractor in combination with the linear echo canceller 3 enables its adaptive

17   filter 7 to operate with a reduced number of delay-line taps. Hence, the

18   amount of computations can be decreased. In the prior art where the linear

19   echo canceller is used exclusively, a reduction of the delay-line taps

20   inevitably results in a significant decrease in the amount of echo that can be

21   cancelled.

22   A modification of the Fourier coefficient subtractors is shown in Fig.

23   5A. In this modification, the Fourier coefficient subtractor additionally

24   includes smoothers 30A and 30B of identical configuration connected

25   between the absolute value circuits 15A, 15B and the divider 16. As a

26   representative, the smoother 30A is shown in detail. It is seen that the

27   smoothers 30A, 30B are identical to the smoother 17 except that their

28   multiplier 22 is supplied with a fixed smoothing constant. The effect of the

29   smoothers 30A and 30B is to smooth out the rapidly varying outputs of the

30   absolute value circuits 15A, 15B, so that the smoother 17 produces a quotient

1   of more stabilized value than that of Fig. 4.

2   In a preferred embodiment shown in Fig. 5B, each of the smoothers

3   30A, 30B includes a smoothing coefficient selector 40 which adaptively

4   controls the scale factor of the multiplier 22 in response to the output of

5   subtractor 21 in a manner similar to the selector 26 of Fig 4. As described

6   previously, when the output of subtractor 21 is positive, the selector 40

7   produces a relatively large smoothing coefficient of 0.01. When the

8   subtractor 21 produces a negative output, the selector 40 supplies a relatively

9   small smoothing coefficient of 0.001 to the multiplier 22. With these time-

10   varying smoothing scale factors, the output of each smoother 30 varies

11   sharply at the rising edge of a transition and varies slowly at the falling edge

12   of the transition. It is known that, in most instances, the amplitude variation,

13   or envelope of human voices and music sounds has a sharply rising edge and

14   a gradually falling edge. Smoothers 30A, 30B of Fig. 5B are best suited for

15   such applications. The operating performance of the Fourier coefficient

16   subtractor of Fig. 5A is further improved with the use of smoothing

17   coefficient selector 40 of Fig. 5B.

18   According to a second embodiment, the spectral shaper 10 is

19   implemented in a spectral suppressor as shown in Fig. 6, in which parts

20   corresponding in significance to those in Fig. 3 are marked with the same

21   numerals and the description thereof is omitted. It will be seen that the

22   spectral suppressor differs from the spectral subtractor of Fig. 3 in that the

23   Fourier coefficient subtractors 13-1 ~ 13-m of Fig. 3 are replaced with Fourier

24   coefficient multipliers 53-1 ~ 53-m.

25   As shown in detail in Fig. 10, each of the Fourier coefficient multipliers

26   53 is comprised of absolute value circuits 60A and 60B, which receive the

27   subband components Si and Ri from the Fourier transform converters 11 and

28   12 and produce absolute values of the Fourier coefficients Si and Ri,

29   respectively. A ratio of the absolute value Si to the absolute value Ri is

30   obtained by divider 61A and a ratio of the absolute value Ri to the absolute

1   value Si is obtained by divider 61B. The output of divider 61A is coupled to a

2   smoother 62 which is identical to the smoother 17 of Fig. 4. The smoother 62

3   produces an average value of Si/Ri ratios with a varying time constant

4   depending on the level of speech activity detected by the speech detector 5.

5   When near-end speech activity is high, the time constant is set to a large or

6   infinite value. When the speech activity is low or zero, the smoother 62

7   operates with a small time constant value. Smoother 62 has the effect of

8   stabilizing the Fourier coefficient multiplier 53 when the ratio varies violently

9   in response to a high near-end speech signal so that a reliable output signal is

10   obtained. The output of smoother 62 is coupled to a multiplier 63 to which

11   the output of divider 61B is also applied. The ratio Ri/Si of divider 61B is

12   multiplied by the smoothed value of ratios Si/Ri from smoother 62 in the

13   multiplier 63 and then smoothed by a smoother 64 of the fixed time-constant

14   type identical to those shown in Figs. 5A, 5B. The output of smoother 64 is

15   fed to a subtractor 65 which subtracts a constant value of "1.0" from the input

16   signal. By comparing the Fourier coefficient multiplier of Fig. 7 with the

17   Fourier coefficient subtractor of Fig. 4, it will be seen that the output signal P4

18   of subtractor 65 is equal to the output signal P3 of subtractor 19, as given by

19   Equation (4a), divided by the near-end signal S and averaged. P4 is

20   represented by:

21        $P4 = Av [P3/S]$

22           $= Av [1 - \{(R/S) \times Av [(E + N)/R]\}]$

23           $= 1 - Av [\{(R/S) \times Av [(E + N)/R]\}]$ (5)

24   Further, the output signal P4 can be obtained by dividing Equation

25   (4b) by S and averaging the result of the division as follows:

26        $P4 = Av [\{(A + E + N) - Ex [E + N]/S]$

27           $= Av [Ex [A]/S]$

28           $= Ex [A/S]$ (6)

29   By comparing Equation (5) to Equation (6), it will be seen that the

30   output signal P4 of subtractor 65 represents an estimate of the amount of

1   near-end talker's speech component contained in a local signal.

2   The input signal Si is then multiplied in a multiplier 66 by the output

3   of subtractor 65 to produce an output signal for application to the inverse

4   Fourier transform converter 14 of Fig. 6. The output signal of this Fourier

5   coefficient multiplier 53i represents an estimate of the Fourier coefficient of

6   echo-suppressed near-end subband speech component.

7   Since the signal P3 of the previous embodiment represents an estimate of the

8   Fourier coefficient of near-end speech component containing no disturbing

9   components including linear echo, noise, and harmonic echo (resulting from

10   the nonlinear characteristics of transducer elements), the signal P4 is also free

11   from the harmonic echo. Therefore, the multiplication of a subband input

12   signal Si by P4 results in the elimination of its harmonic echo component.

13   Similar to the previous embodiment, the spectral suppressor can remove a

14   residual echo resulting from the echo canceller 3 making a false echo path

15   estimation.

16   As illustrated in Fig. 8, each Fourier coefficient multiplier 53 is

17   preferably provided with smoothers 67A and 67B between the absolute value

18   circuits 60A, 60B and the dividers 61A, 61B. The effect of the smoothers 67A

19   and 67B is to smooth out the rapidly varying outputs of the absolute value

20   circuits 60A, 60B, so that the smoother 62 produces a quotient of more

21   stabilized value than that of Fig. 7.

22   Fig. 9 illustrates a modification of the previous embodiments. In this

23   modification, a harmonics generator 70 is provided between the output of

24   linear echo canceller 3 and the spectral shaper 10. Harmonics generator 70

25   emphasizes the harmonic components of an echo replica generated by the

26   echo canceller 3 so that the echo replica supplied to the spectral shaper 10

27   contains a harmonics replica of the distant signal. This harmonics replica is

28   similar to the nonlinear distortion of the acoustic echo path. The provision of

29   the harmonics generator 70 enables the spectral shaper 10 to perform its echo

30   suppression in an efficient manner.

1   While the spectral shaper 10 has been described as operating on a

2 . sample-by-sample basis to perform its Fourier transformation, the amount of

3   computations can be reduced by performing Fourier transform on a frame-

4   by-frame basis using the overlap save and overlap add techniques as

5   described in a paper "Frequency-Domain and Multirate Adaptive Filtering",

6   John J. Shynk, IEEE signal Processing Magazine, January 1992, pages 14-37.

7   Linear echo cancellers that perform echo cancellation in a linear

8   transform domain and reconstitution in an inverse transform domain is

9   called a transform-domain echo canceller. The amount of computations can

10   be reduced by implementing the linear echo canceller 3 in a transform-

11   domain echo canceller configuration and implementing the spectral shaper 10

12   in a transform domain configuration identical to that of the echo canceller.

13   An echo suppressor incorporating such a transform-domain echo

14   canceller 80 and a transform-domain spectral shaper 81 is shown in Fig. 10.

15   As illustrated, the echo canceller 80 generates and supplies a first set of

16   transform-domain subband frequency components S1 ~ Sm and a second set

17   of transform-domain subband frequency components R1 ~ Rm to the

18   spectral subtractor 81.

19   As shown in detail in Fig. 11, the echo canceller 80 is comprised of a

20   Fourier transform converter 90 in which the distant signal is converted to a

21   set of transform-domain subband frequency components and supplied to

22   corresponding adaptive filters of an adaptive filter bank 91. The output of

23   subtractor 4 is also converted in a Fourier transform converter 92 to a set of

24   transform-domain subband frequency components S1 ~ Sm and supplied to

25   corresponding multipliers 93, whose outputs are supplied to the adaptive

26   filter bank 91. Adaptive filter bank 91 produces and supplies a set of

27   transform-domain subband frequency components R1 ~ Rm to an inverse

28   Fourier transform converter 94 where the input signals are inversely

29   processed and combined together into an output signal for coupling to the

30   subtractor 4 as well as to the spectral detector 5.

1   Multipliers 93 are controlled by the speech detector 5 so that their

2   outputs are forced to zero when the near-end speech activity is high.

3   To the spectral subtractor 81 a first set of transform-domain subband

4   components S1 ~ Sm and a second set of transform-domain subband

5   components R1 ~ Rm are supplied from the Fourier transform converter 92

6   and the adaptive filter bank 91, respectively.

7   Spectral subtractor 81 includes a plurality of Fourier coefficient

8   subtractors 100-1 ~ 100-m, as shown in Fig. 12. Similar to that shown in Fig.

9   3, each Fourier coefficient subtractor 100-i has a pair of input terminals for

10   receiving and processing signals Si and Ri from the echo canceller 80

11   according to the output of the speech detector 5 and feeds its output to an

12   inverse Fourier transform converter 101.

13   A third embodiment of the present invention is shown in Fig. 13. In

14   this embodiment, a Fourier transform converter 111 converts the output of

15   microphone 1 into a first set of subband component signals S1 ~ Sm and a

16   Fourier transform converter 112 converts a copy of the distant signal supplied

17   to the loudspeaker 2 into a second set of subband component signals R1 ~

18   Rm. A plurality of subband echo suppressors 113 of identical structure are

19   provided corresponding in number to the subband frequency component

20   signals of each set. An inverse Fourier transform converter 114 is provided to

21   combine the output signals of the subband echo suppressors 113 into a local

22   signal.

23   Each subband echo suppressor 113 is basically of the same

24   configuration as that described previously. It includes an echo canceller 115,

25   a subtractor 116, a speech detector 117, but differs from the previous

26   embodiments in that the spectral shaper 10 is replaced with a Fourier

27   coefficient subtractor 118 of the type previously shown in Figs. 4, 5A and 5B

28   (or Fourier coefficient multiplier of Figs. 7 and 8). Echo canceller 115 receives

29   and processes a corresponding one of the outputs of Fourier transform

30   converter 112 for feeding the subtractor 116. Subtractor 116 receives a

1 corresponding one of the outputs of the Fourier transform converter 111 for

2 feeding the Fourier coefficient subtractor 118.

3   While mention has been made of Fourier transform, linear transform

4 such as discrete cosine transform and filter-bank transform can equally be

5 used as well.